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(71) Applicant: ERICSSON INC. [US/US]; 7001 Development Drive, Research Triangle Park, NC 27709 (US).

(72) Inventor: IRVIN, David, R.; 1546 Ire Dell Drive, Raleigh, NC 27608 (US).

(74) Agent: MONCO, Dean, A.; Wood, Phillips, VanSanten, Clark & Mortimer, Suite 3800, 500 West Madison Street, Chicago, IL 60661-2511 (US).

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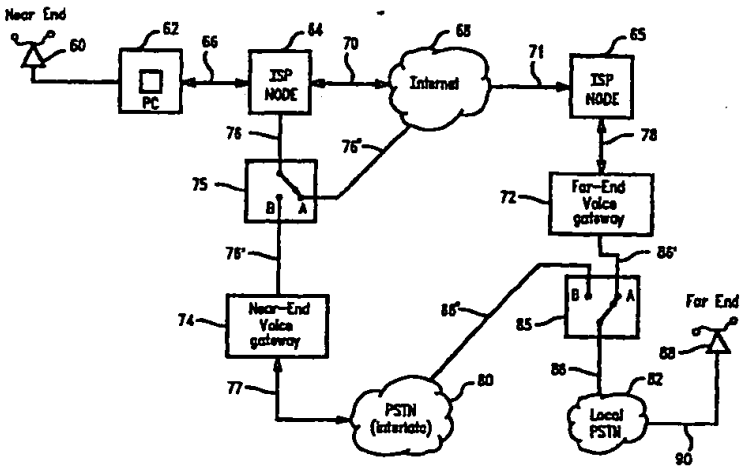
(54) Title: QUALITY-OF-SERVICE BASED TELEPHONY OVER INTERNET VIA PAIRED VOICE GATEWAYS



(57) Abstract: A telephony session between a near-end calling party and a far-end calling party is established through a near-end voice gateway and a far-end voice gateway within a telephone communication system. A subscriber to the communication system chooses, either dynamically when the subscriber places a call set-up request or when the subscriber signs up for services with an Internet Service Provider, a high quality service, a low cost service, or an opportunistic service, and is charged accordingly. When a high quality service is chosen, all of the subscriber's calls are routed via a near-end voice gateway. When a low cost service is chosen, all of the subscriber's calls are routed via a far-end voice gateway. When opportunistic service is chosen, calls are routed via either the far-end voice gateway or the near-end voice gateway. The gateway selected depends on the Internet's capability to meet a quality of service (QoS) level specified by the subscriber. When the performance of the Internet can support the specified QoS, the call is routed via the far-end voice gateway. When the Internet cannot support the specified QoS level, the call is routed via the near-end voice gateway.

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QUALITY-OF-SERVICE BASED TELEPHONY OVER INTERNET VIA PAIRED VOICE GATEWAYS

BACKGROUND OF THE INVENTION

The present invention relates generally to voice telephony sessions using voice-over-Internet protocols (VOIP), and more particularly, to a method and apparatus for using VOIP techniques to connect a near-end calling party to a far-end called party selectively through a circuit-switched network using a near-end voice gateway or through a packet-switched network using a far-end voice gateway.

Voice communication over the Internet via Internet protocols (IPs) has evolved into a market with a number of applications such as voice telephony, voice-mail, fax, desktop video-conferencing, application sharing and document sharing. IP telephony can be achieved, at least in principle, on any data network that uses IP, such as the Internet or any of a growing number of Intranets and Local Area Networks (LANs).

Telephony carried over IP networks has generated much attention for end users, Internet Service Providers (ISPs), data vendors, telephony vendors, and existing telecom operators because the IP networks provide telephony services at a very low price, in comparison to the traditional circuit-switched networks, i.e., the Public Switched Telephone Network (PSTN). By routing voice traffic over IP networks, users can avoid long-distance charges completely, paying only for a local connection.

IP telephony is quite different from traditional circuit-switched telephony, where an end-to-end circuit is set up between two telephones. For example, a circuit-switched telephony connection is established for the duration of each telephone call, generally via Basic Rate Interface (BRI) or Primary Rate Interface (PRI), with a fixed bandwidth (64 kbit/s) reserved to transmit digitized voice, even during silent moments of a telephone conversation.

IP networks, however, do not maintain a circuit for the duration of the call in the same way as the traditional circuit-switched networks. Rather, the analog voice signal is digitized, compressed and converted into IP datagrams or packets that are transmitted over the IP network and mixed with other IP traffic. IP packets are commonly known in the art and thus will not be further described herein.

Typically, IP telephony "traffic" is handled by a voice gateway that is usually placed between the telephone network, i.e., PSTN, and the IP network. The voice gateway provides the interface between the telephone network and the IP network. The voice gateway may handle the signaling to and from the telephone network, reception of telephone numbers, conversion between telephone numbers and IP addresses in the IP network, as well as voice processing. The voice processing includes reception of the voice signal, compression and packetization, echo cancellation, silence suppression, etc. The voice gateway compresses the voice signal to reduce the amount of bandwidth required in order to reduce cost and to reduce the delay impact from the IP network.

For example, users dial into the Internet Service Provider (ISP) often by using a personal computer equipped for internet telephony. The ISP responds by requesting the user's destination telephone number. A routing table identifies which voice gateway is located closest to the destination telephone network. The IP address of that voice gateway is then used to route the telephone call as datagrams through the IP network. The voice gateway also operates in reverse for packets coming from a far-end called party and going to a near-end calling party. Both operations (coming from and going to the calling party) can take place at the same time, allowing a full-duplex conversation.

FIG. 1 illustrates an example of a telephony communication system using a near-end voice gateway 20. The near-end voice gateway 20 interfaces with an ISP NODE 22 via a communication line 27. The ISP NODE 22 is connected to a telephone 24 and/or a computer 26 of a near-end calling party via a communication line 21. The ISP NODE 22 provides access for the telephone 24 and the computer 26 to a packet-switched network, shown as an Internet 34, via a communication line 25. The near-end voice gateway 20 is connected to a conventional circuit-switched network, shown as a PSTN 30 (for the interlata portion) and a local PSTN 32 via a communication line 29. A telephone 28 of a far-end called party is connected to the local PSTN 32 via a communication line 31. The local PSTN 32 is connected to the PSTN 30 via a trunk line 33. The near-end voice gateway 20 receives VOIP packets from the near-end calling party. The near-end voice gateway 20 converts the voice traffic carried by the VOIP packets to a Pulse Code Modulation (PCM) data stream and sends this data stream to the telephone 28 via the PSTN 30 and the local PSTN 32.

The communication system depicted in FIG. 1 provides high quality telephony service because it terminates the use of VOIP packets close to the near-end calling party. Thus, the quality of service (QoS) of the resulting communication path approaches that which is provided by a telephone connection that is essentially conventional (circuit-switched) end-to-end, and yet requires only a single local-loop to serve both the computer 26 and the telephone 24.

A system according to FIG. 1 can be provided by the Ericsson Phone Doubler, which uses a near-end voice gateway that allows a user who is connected to the Internet to make and receive telephone calls without logging off the Internet. The Phone Doubler is described by Allan Hansson, Robert Nedjeral and Ingmar Tönnby, in *Phone Doubler - A step towards integrated Internet and telephone communities*, Ericsson Review, No. 4, pp. 142-151 (1997).

FIG. 2 illustrates an example of a telephony communication system using a far-end voice gateway 40. A first ISP NODE 42 and a second ISP NODE 50 are connected to a packet-switched network, shown as Internet 48, via communication lines 49 and 51, respectively. The far-end voice gateway 40 interfaces with the first ISP NODE 42 and a conventional circuit-switched network, which is illustrated as a local PSTN 54, via communication lines 57 and 55, respectively. A telephone 52 of a far-end called party is connected to the local PSTN 54 via a communication line 53. The second ISP 50 is connected to a telephone 44 and/or a computer 46 of a near-end calling party via a communication line 47.

The near-end calling party transmits voice packets to the Internet 48 via the telephone 44 or the computer 46 and the second ISP NODE 50. The far-end voice gateway 40 receives VOIP packets of the near-end calling party from the Internet 48 via the first ISP NODE 42, converts the VOIP packets to a PCM data stream, and sends this PCM data stream to the telephone 52 via the local PSTN 54. In effect, the far-end voice gateway 40 terminates the use of VOIP packets from the near-end calling party so that toll charges are minimized. Unfortunately, routing a call via the Internet in this way provides highly variable telephony performance, particularly in the Internet's capability to deliver a stream of packets that arrive timely and regularly. Thus, the resulting telephone connection, even though providing a low cost, provides a highly variable QoS level, which can sometimes be quite poor.

Since establishing a telephony session via either a near-end voice gateway as depicted in FIG. 1 or a far-end voice gateway as depicted in FIG. 2 affects both the price users pay for the telephony session and the quality of the telephony session, there is a need for a method and apparatus that provides users with a choice of routing a call exclusively through a circuit-switched network using a near-end voice gateway or routing the call through a packet-switched network using a far-end voice gateway. There is a further need for a method and apparatus that allows a user to select a desired communication path or to specify a QoS, either dynamically at the time the user places each call or when the user signs up for service with an ISP. There is a still further need to gather information regarding the performance of the packet-switched network so that calls may be placed in the most economic way that meets the user's QoS expectations.

SUMMARY OF THE INVENTION

According to the present invention, these and other needs are met by methods and apparatus for selectively establishing telephony sessions through circuit-switched networks via near-end voice gateways or packet-switched networks via far-end voice gateways.

According to an exemplary embodiment of the present invention, a communication system has both a near-end voice gateway and a far-end voice gateway. A subscriber of the communication system specifies a class of service. For example, the subscriber may select a high quality service, low cost service, or opportunistic service (a hybrid of high quality and low cost) and be billed accordingly. The subscriber may specify the class of service either dynamically at the time a call is placed or when the subscriber signs up for service with an ISP.

When the high quality service is chosen, all of the subscriber's calls (or a particular call if the selection is dynamic at the subscriber's option) are routed through the near-end voice gateway. When the low cost service is chosen, all of the subscriber's calls (or a particular call if the selection is dynamic at the subscriber's option) are routed through the far-end voice gateway. When opportunistic service is chosen, the subscriber's calls are selectively routed through either the far-end voice gateway or the near-end voice gateway. The gateway selected depends on the Internet's capability to meet a QoS specified by the subscriber. When the Internet can meet the subscriber's specified QoS, the call is routed through the far-end voice gateway. When the Internet cannot meet the subscriber's specified QoS, the call is routed

through the near-end voice gateway. The subscriber is billed according to the path actually used.

BRIEF DESCRIPTION OF THE DRAWINGS

The features and advantages of the present invention will become apparent by reading this description in conjunction with the accompanying drawings, in which like reference numerals refer to like elements and in which:

FIG. 1 is a block diagram that illustrates a prior art telephony communication system using a near-end voice gateway;

FIG. 2 is a block diagram that illustrates a prior art telephony communication system using a far-end voice gateway;

FIG. 3 is a block diagram that illustrates a telephony communication system using both a near-end voice gateway and a far-end voice gateway according to an exemplary embodiment of the present invention;

FIG. 4 is a flow chart that illustrates a method of establishing a telephony session according to an exemplary embodiment of the present invention;

FIG. 5 is a sub-flow chart that illustrates a method of determining the performance of a packet-switched network according to an exemplary embodiment of the present invention; and

FIG. 6 is a sub-flow chart that illustrates another method of determining the performance of a packet-switched network according to an exemplary embodiment of the present invention.

DETAILED DESCRIPTION OF THE INVENTION

FIG. 3 illustrates a telephony communication system according to an exemplary embodiment of the present invention. In FIG. 3, the telephony communication system utilizes both a near-end voice gateway 74 and a far-end voice gateway 72. At the near-end calling party side, a telephone 60 and/or a computer 62 are coupled to a first ISP NODE 64 via a communication line 66. The telephone 60 or the computer 62 may serve as the near-end calling party's telephone unit. At the far-end called party side, a telephone 88 is connected to a local PSTN 82 via a communication line 90. The near-end voice gateway 74 and the far-end

voice gateway 72 connect the first ISP NODE 64 and a second ISP NODE 65 to a circuit-switched network, illustrated as a PSTN 80 (for the interlata portion, if any) and the local PSTN 82, respectively. The first and second ISP NODES 64 and 65 are connected to a packet-switched network, illustrated as an Internet 68 via communication lines 70 and 71, respectively. The PSTN 80 is connected to the near-end voice gateway 74 via a communication line 77. A switch 75 is connected to the first ISP 64 via a communication line 76. The switch 75 has an A terminal and a B terminal that are connected to the Internet 68 and the near-end voice gateway 74, via communication lines 76" and 76', respectively. A switch 85 is connected to the local PSTN 82 via a communication line 86. The switch 85 has an A terminal and a B terminal that are connected to the far-end voice gateway 72 and the PSTN 80 via communication lines 86' and 86", respectively.

The switches 75 and 85 are symbolic to show how traffic is routed; the functions of the switches 75 and 85 are carried out by conventional message addressing and call setup signaling. When a telephony session is connected between the near-end calling party and the far-end called party via the near-end voice gateway 74, traffic flows as though the symbolic switches 75 and 85 routed the call through their respective B terminals. When a telephony session is connected between the near-end calling party and the far-end called party via the far-end voice gateway, traffic flows as though the symbolic switches 75 and 85 routed the call via their respective A terminals.

The communication lines described in FIG. 3 may all be plain old telephone service (POTS) communication lines, although other types of communication lines may well be used. For example, communication line 66 may be an integrated service digital network (ISDN) line, whereas communication line 70 may be a T1 (1.533 Mbit/s) or an E1 (2.0488 Mbit/s) trunk line.

Although the exemplary embodiment described in FIG. 3 illustrates a single near-end calling party served by a single near-end voice gateway 74, and a single far-end called party served by a single far-end voice gateway 72, it will be appreciated by those of ordinary skill in the art that many users may be connected to the communication system. It will also be appreciated by those of ordinary skill in the art that the present invention may have many different embodiments. For example, any number of Internets or LANs may be used as the packet-switched network 68. The near-end and far-end voice gateways also may both be

served by a single ISP or by a plurality of ISPs. Telephone servers or Private Branch Exchanges (PBXs) may also be used to implement the invention.

According to the invention, a subscriber, e.g., a near-end calling party, can choose to establish a telephony session via either a near-end voice gateway or a far-end voice gateway based a class of service specified by the subscriber to an ISP that is serving the subscriber. The class of service specified by the subscriber, for example, may be high quality service, in which the call would be transmitted via the circuit-switched network, low cost service, in which the call would be transmitted via a packet-switched network, or opportunistic service (a hybrid of high quality and low cost), choosing between the near-end voice gateway or the far-end voice gateway based on the performance level of the Internet. The subscriber can specify the class of service, either dynamically at the time the subscriber places a call or when the subscriber signs up for service with the ISP, in a manner that will be described later in connection with FIG. 4.

Referring to FIG. 3, for illustrative purposes, once a subscriber signs up for service with an ISP, the subscriber can authorize the first ISP NODE 64 to establish its telephony sessions via either the near-end voice gateway 74 (high quality service) or the far-end voice gateway 72 (low cost service). Alternatively, the subscriber may sign up for a service that allows the subscriber to selectively establish a call via either the near-end voice gateway 74 or the far-end voice gateway 72, i.e., opportunistic service. In this approach, the subscriber may send explicit signaling information to the first ISP NODE 64 in a call setup request identifying whether to route the call via the near-end voice gateway 74 or the far-end voice gateway 72.

When the subscriber selects the high quality service, the subscriber's calls may be routed via the near-end voice gateway 74. For example, all of the subscriber's calls (or a particular call if the selection is dynamic at the subscriber's option) would be converted from VOIP packets to circuit-switched PCM by the near-end voice gateway 74, and routed to the telephone 88 from the near-end voice gateway 74 via the PSTN 80 and the local PSTN 82. When the subscriber selects low cost service, the subscriber's calls would be routed via the far-end voice gateway 72. For example, all of the subscriber's calls (or a particular call if selection would be dynamic at the subscriber's option) would be converted from VOIP packets to circuit-switched PCM by the far-end voice gateway 72. The communication path in this case

is through the first ISP NODE 64, the Internet 68, the second ISP NODE 65, the far-end voice gateway 72 and the local PSTN 82.

When the subscriber selects opportunistic service, the subscriber's calls would be selectively routed via either the far-end voice gateway 72 or the near-end voice gateway 74. In this approach, the gateway selected by the first ISP NODE 64 depends on a QoS specified by the subscriber and the ability of the Internet to meet the specified QoS. For example, the performance of the Internet 68, e.g., the Internet's ability to transmit packets when the subscriber places the call setup request to the first ISP NODE 64, will be evaluated and compared to the subscriber's specified QoS. Illustrative ways of determining the performance of the Internet 68 will be described later in connection with FIGS. 5 and 6.

With opportunistic service, the subscriber, for example, will specify an acceptable QoS level to the first ISP NODE 64 either at the time the subscriber signs up for service with the ISP or when the subscriber places the call setup request. The subscriber's acceptable QoS level, for example, may be based on transmission delay of the Internet or other parameters described below. In this approach, when the performance of the Internet 68 is adequate to provide service that meets the subscriber's specified QoS level, the subscriber's calls may be routed via the Internet 68. Thus, the VOIP packet to circuit-switched PCM data stream conversion takes place at the far-end voice gateway 72. The resulting PCM data stream is routed to the telephone 88 via the local PSTN 82. When the performance of the Internet 68 cannot support the subscriber's specified QoS level, the VOIP packet to circuit-switched PCM data stream conversion takes place at the near-end voice gateway 74. The resulting PCM data stream is routed from the near-end voice gateway 74 to the telephone 88 via both PSTN 80 and local PSTN 82.

FIG. 4 is a flow chart that illustrates a method of a call-routing aspect according to an exemplary embodiment of the present invention. In FIG. 4, the method begins at step 400 where an ISP receives a call setup request from a subscriber. In step 405, the ISP determines the subscriber's specified class of service as previously described. The class of service desired by the subscriber may be determined by the subscriber when the subscriber signs up for service with the ISP or the call setup request sent to the ISP by the subscriber may contain explicit signaling information regarding the specified class of service. Next at step 408, the ISP selects the subscriber's specified class of service, as determined in step 405.

When the subscriber requests, for example, high quality service, the ISP selects the near-end voice gateway to route the call via the circuit-switched network (or other high quality network), for example, using the methods of Phone Doubler as previously described or other methods known in the art (step 410). The ISP notes the selected service in appropriate call detail records for billing purposes. When the subscriber requests, for example, low cost service, the ISP selects the far-end voice gateway to establish a VOIP call that terminates at the far-end voice gateway (step 415). The ISP notes the action in appropriate call detail records for billing purposes.

When opportunistic service is requested by the subscriber, the ISP selectively routes the call through the circuit-switched network using the near-end voice gateway or the packet-switched network using the far-end voice gateway, based on whether the performance of the Internet meets the subscriber's specified QoS, when the subscriber places the call setup request to the ISP. Referring back to step 408 of FIG. 4, if the subscriber signed up for opportunistic service or signals the use of opportunistic service during call set up, the method proceeds to step 420, where the ISP determines the subscriber's specified QoS for choosing whether to route the call via the circuit-switched network or the packet-switched network. The subscriber's specified QoS may be based on, for example, a predetermined threshold coefficient of variation as will be described later in connection with FIG. 6.

At step 425, the ISP determines the performance level of the Internet in response to the subscriber's placing the call setup request, which will be compared to the subscriber's specified QoS from step 420 to determine whether to route the call via the near-end voice gateway or the far-end voice gateway. The present invention anticipates a number of ways of determining the performance level of the Internet, for example, round-trip delay, one-way delay, variation in round-trip delay or one-way delay, likelihood of call initiation or call completion, available speech quality as limited by available bandwidth and coding rate. Illustrative ways of determining the performance level of the Internet will be described later in connection with FIGS. 5 and 6.

Still referring to FIG. 4, at step 430, if the performance level of the Internet meets or exceeds the subscriber's specified QoS as determined in step 420, the method proceeds to step 415, where the ISP selects the far-end voice gateway to route the call via the Internet, as described above. The ISP notes the action in appropriate call detail records for billing

purposes. If the performance level of the Internet does not meet the subscriber's specified QoS as determined in step 420, the method proceeds to step 410 where the ISP selects the near-end voice gateway to route the call via the circuit-switched network (or other high quality network) using the methods of Phone Doubler or other appropriate methods. The ISP notes the action in appropriate call detail records for billing purposes.

FIG. 5 shows a preferred way of determining the performance of the Internet. As illustrated in FIG. 5, the method begins at step 500, where the near-end voice gateway generates a reconnaissance or wrap-back packet, based on instructions from the ISP. At step 505, the near-end voice gateway sends the reconnaissance or wrap-back packet to the far-end voice gateway via the Internet and records the departure time of the packet. Next at step 510, the near-end voice gateway awaits the return of the reconnaissance or wrap-back packet from the Internet. For example, the near-end voice gateway sends a packet via the Internet to the far-end voice gateway. The far-end voice gateway sends the packet back to the near-end voice gateway via the Internet.

Next at step 515, the near-end voice gateway computes the round-trip delay by subtracting the transmission time of the packet from the time the packet returns to the near-end voice gateway. The near-end voice gateway transmits the computed round-trip delay to the ISP. Based on the round-trip delay, the performance level of the Internet can be determined. For example, a small round-trip delay means that the packet was transmitted quickly through the Internet. Thus, it is reasonable to assume that the performance of the Internet will support a high QoS level, and that the Internet would be able to transmit packets between the near-end calling party and the far-end called party without unreasonable delay. It will be appreciated by those of ordinary skill in the art that several such round-trip delays may be computed according to steps 500-515 (e.g., several wrap-back packets may be sent consecutively or from time to time). The round-trip delays may be averaged, or the median round-trip delay found, or the largest and smallest round-trip delays combined or eliminated from consideration, to determine other estimates of Internet performance. It will also be appreciated that other network latency estimates may be used, for example one-way delay, as well as round-trip delay.

Alternatively, the performance level of the Internet may be determined by computing a coefficient of variation, as illustrated in FIG. 6. According to FIG. 6, the method begins at

step 600 where the ISP instructs the near-end voice gateway to find one or more calls in progress between the near-end voice gateway and the far-end voice gateway. The near-end voice gateway recalls data from a memory regarding the time-sampled occupancy over a predetermined interval of recent history of packet reassembly buffers in the near-end voice gateway or, remotely, in the far-end voice gateway, that supports those calls in progress (step 605). When the reassembly buffers cannot process a steady flow of data, the performance of the Internet is judged to be poor, meaning that the round-trip delay is highly variable, and therefore the subscriber's QoS requirements cannot be met.

The reassembly buffer, for example, can be thought of as a First-In-First-Out (FIFO) queue, whose queue length is recorded either upon arrival/departure of a byte or at regular intervals. This sampling, for example, gives rise to a time series:

Time	Queue length
t_1	l_1
t_2	l_2
t_n	l_n

The time series is truncated (not allowed to grow forever) by a sliding window, i.e., $t_1 \ldots t_n$ is replaced by $t_2 \ldots t_{n+1}$, which is replaced by $t_3 \ldots t_{n+2}$ and so on. These $t_n \ldots t_j$ are the samples are the numbers from which the coefficient of variation is computed. The reconstruction of signals is described in the article by Po L. Tein and Maria C. Yuang, "Intelligent Voice Smoother for Silence-Suppressed Voice Over Internet," IEEE J. on Selected Areas In Communications, vol. 1, pp. 29-41, (Jan. 1999).

At step 610, the mean and standard deviation of the time samples of buffer occupancy are computed, and from these results, the coefficient of variation (standard deviation divided by mean) is computed. The computed coefficient of variation is compared to a predetermined threshold coefficient of variation at step 615. The predetermined threshold coefficient of variation is determined by the subscriber to correspond to an acceptable QoS level. At step 618, if the computed coefficient of variation exceeds the predetermined threshold coefficient of variation, the variability of the Internet is believed to be excessive, e.g., the performance level of the Internet might be poor. Consequently, the ISP establishes the call via the near-end voice gateway, which transmits the call to the circuit-switched network (or other high quality

network) using the methods of Phone Doubler or other appropriate methods, and the ISP notes the action in appropriate call detail records for billing purposes (step 620). If the computed coefficient of variation does not exceed the predetermined threshold coefficient of variation, the ISP routes the call via the far-end voice gateway (step 625). The ISP notes the action in appropriate call detail records for billing purposes.

It will also be appreciated by those of ordinary skill in the art that the decision to terminate use of the Internet at the near-end voice gateway rather than the far-end voice gateway can be based on a particular Internet route for transmitting a call and/or the time of day the call is made. For example, certain Internet routes are congested during certain times of a day, and therefore should be avoided during those times; these Internet routes are not congested during other times of the day. Further, the near-end voice gateway and the far-end voice gateway may exchange information concerning the performance observed from other calls in progress between the near-end voice gateway and the far-end voice gateway, and make decisions whether to route a call via the Internet accordingly. Also, a compound criteria can be used to determine the performance of the Internet. For example, results of a round-trip delay as described in FIG. 5 and a coefficient of variation as described in FIG. 6 may both be evaluated to determine the performance of the Internet.

It will be appreciated by those of ordinary skill in the art that the present invention can be embodied in other forms without departing from its essential character. For example, although the present invention has been described in terms of wireline networks, any portion(s) of the networks or links involved may be provided by cellular telephony or other wireless means for voice or data transmission.

CLAIMS

WHAT IS CLAIMED IS:

1. A method comprising:

generating a call setup request,

determining a class of service requirement based on the call setup request, and

establishing a telephony session between a near-end calling party and a far-end calling party based on the class of service requirement.

2. The method of claim 1, wherein establishing a telephony session between a near-end calling party and a far-end calling party based on the class of service requirement comprises establishing the telephony session through a circuit-switched network if the class of service requirement is a high quality service.

3. The method of claim 2, wherein establishing the telephony session through a circuit-switched network if the class of service requirement is a high quality service comprises establishing the telephony session through the circuit-switched network using a near-end voice gateway if the class of service requirement is the high quality service.

4. The method of claim 1, wherein establishing a telephony session between a near-end calling party and a far-end calling party based on the class of service requirement comprises establishing the telephony session through a packet-switched network if the class of service requirement is a low cost service.

5. The method of claim 4, wherein establishing the telephony session through a packet-switched network if the class of service requirement is a low cost service comprises establishing the telephony session through the packet-switched network using a far-end voice gateway if the class of service requirement is the low cost service.

6. The method of claim 4, wherein the packet-switched network is an Internet.

7. The method of claim 4, wherein the packet-switched network is an Intranet.

8. The method of claim 1, further comprising determining a Quality of Service requirement, and wherein establishing a telephony session between a near-end calling party and a far-end calling party based on the class of service requirement comprises establishing the telephony session through a circuit-switched network or a packet-switch network in response to the Quality of Service requirement if said class of service requirement is an opportunistic service.

9. The method of claim 1, wherein the class of service requirement is selected from a high quality service, a low cost service or an opportunistic service.

10. The method of claim 1, further comprising billing a subscriber according to the class of service requirement.

11. The method of claim 1, wherein the class of service requirement is dynamically selected when a subscriber places the call set-up request.

12. The method of claim 1, wherein the class of service requirement is selected by a subscriber when the subscriber signs up for services with an Internet Service Provider.

13. A method comprising:

generating a call setup request,

determining a class of service requirement based on the call setup request, and

establishing the telephony session between the near-end calling party and the far-end called party selectively through a near-end voice gateway or a far-end voice gateway, based on the class of service requirement.

14. The method of claim 13, wherein establishing a telephony session between a near-end calling party and a far-end calling party through a near-end voice gateway or a far-end voice gateway, based on the class of service requirement comprises establishing the telephony session through a circuit-switched network if the class of service requirement is a high quality service.

15. The method of claim 13, wherein establishing a telephony session between a near-end calling party and a far-end calling party through a near-end voice gateway or a far-end voice gateway, based on the class of service requirement comprises establishing the telephony session through a packet-switched network if the class of service requirement is a low cost service.

16. The method of claim 15, wherein the packet-switched network is an Internet.

17. The method of claim 15, wherein the packet-switched network is an Intranet.

18. The method of claim 13, further comprising determining a Quality of Service requirement, and wherein establishing a telephony session between a near-end calling party and a far-end calling party through a near-end voice gateway or a far-end voice gateway, based on the class of service requirement comprises establishing the telephony session through a circuit-switched network or a packet-switch network in response to the Quality of Service requirement if said class of service requirement is an opportunistic service.

19. The method of claim 13, wherein the class of service requirement is selected from a high quality service, a low cost service or an opportunistic service.

20. The method of claim 13, further comprising billing a subscriber according to the class of service requirement.

21. The method of claim 13, further comprising determining a performance of a packet-switched network.

22. The method of claim 21, wherein establishing a telephony session between a near-end calling party and a far-end calling party through a near-end voice gateway or a far-end voice gateway, based on the class of service requirement further comprises establishing the telephony session via the near-end voice gateway or the far-end voice gateway based on the performance of the packet-switched network.

23. The method of claim 22, further comprising computing a round-trip delay to determine the performance of the packet-switched network.

24. The method of claim 23, wherein computing the round-trip delay further comprises:

generating a reconnaissance or wrap-back packet,

transmitting the reconnaissance or wrap-back packet via the packet-switched network,

awaiting return of the reconnaissance or wrap-back packet from the packet-switched network, and

calculating a network latency estimate upon return of the reconnaissance or wrap-back packet.

25. The method of claim 24, further comprising recording the time the reconnaissance or wrap-back packet is sent via the packet-switched network, and recording the time the reconnaissance or wrap-back packet returns from the packet-switched network, and computing the network latency estimate therefrom.

26. The method of claim 24, wherein generating a reconnaissance or wrap-back packet, transmitting the reconnaissance or wrap-back packet via the packet-switched network, awaiting return of the reconnaissance or wrap-back packet from the packet-switched network, and calculating a network latency estimate upon return of the reconnaissance or wrap-back packet are repeated at least once to compute a plurality of network latency estimates, and further comprising finding an average value of the plurality of network latency estimates.

27. The method of claim 24, wherein generating a reconnaissance or wrap-back packet, transmitting the reconnaissance or wrap-back packet via the packet-switched network, awaiting return of the reconnaissance or wrap-back packet from the packet-switched network, and calculating a network latency estimate upon return of the reconnaissance or wrap-back packet are repeated at least once to compute a plurality of network latency estimates, and further comprising finding a median value of the plurality of network latency estimates.

28. The method of claim 22, further comprising:

locating telephone calls in progress between the near-end voice gateway and the far-end voice gateway,

recalling data from a memory regarding the telephone calls in progress,

computing a coefficient of variation based on the data,

comparing the calculated coefficient of variation to a predetermined threshold coefficient of variation, and

determining whether to establish the telephony session via the packet-switched network or a circuit-switched network based on results of comparing the calculated coefficient of variation to the predetermined threshold coefficient of variation.

29. The method of claim 28, wherein establishing the telephony session via the near-end voice gateway or the far-end voice gateway based on the performance of the packet-switched network further comprises establishing the telephony session using the packet-switched network provided that the calculated coefficient of variation is equal to or less than the predetermined threshold coefficient of variation.

30. The method of claim 28, wherein establishing the telephony session via a near-end voice gateway or a far-end voice gateway based on the performance of the packet-switched network further comprises establishing the telephony session using the circuit-switched network provided that the calculated coefficient of variation is greater than the predetermined threshold coefficient of variation.

31. The method of claim 22, wherein determining the performance of the packet-switched network is based on a particular packet-switched network route and a time of day the telephony session is requested.

32. The method of claim 22, wherein determining the performance of the packet-switched network further comprises exchanging reports between the near-end voice gateway and the far-end voice gateway concerning the performance of other calls in progress between the near-end voice gateway and the far-end voice gateway.

33. The method of claim 22, further comprising utilizing both a network latency estimate and a coefficient of variation to determine the performance of the packet-switched network.

34. The method of claim 13, wherein the class of service requirement is dynamically selected when a subscriber places the call set-up request.

35. The method of claim 13, wherein the class of service requirement is selected by a subscriber when the subscriber signs up for services with an Internet Service Provider.

36. A communication system comprising:

a first telephone unit connected to a first Internet Service Provider (ISP) NODE, the first ISP NODE connected both to a packet-switched network and a near-end voice gateway, the first ISP NODE providing access for the first telephone unit to the packet-switched network and to the near-end voice gateway,

a second telephone unit connected to a circuit-switched network, the circuit-switched network connected to both the near-end voice gateway and a far-end voice gateway, and

a second ISP NODE connected to both the packet-switched network and the far-end voice gateway, the second ISP NODE providing access for the second telephone unit to the packet-switched network,

wherein the communication system selectively establishes a telephony session between a near-end calling party and a far-end calling party through the near-end voice gateway or the far-end voice gateway, based on a class of service.

37. The communication system of claim 36, wherein the class of service is a high quality service.

38. The communication system of claim 36, wherein the class of service is a low cost service.

39. The communication system of claim 36, wherein the class of service is an opportunistic service.

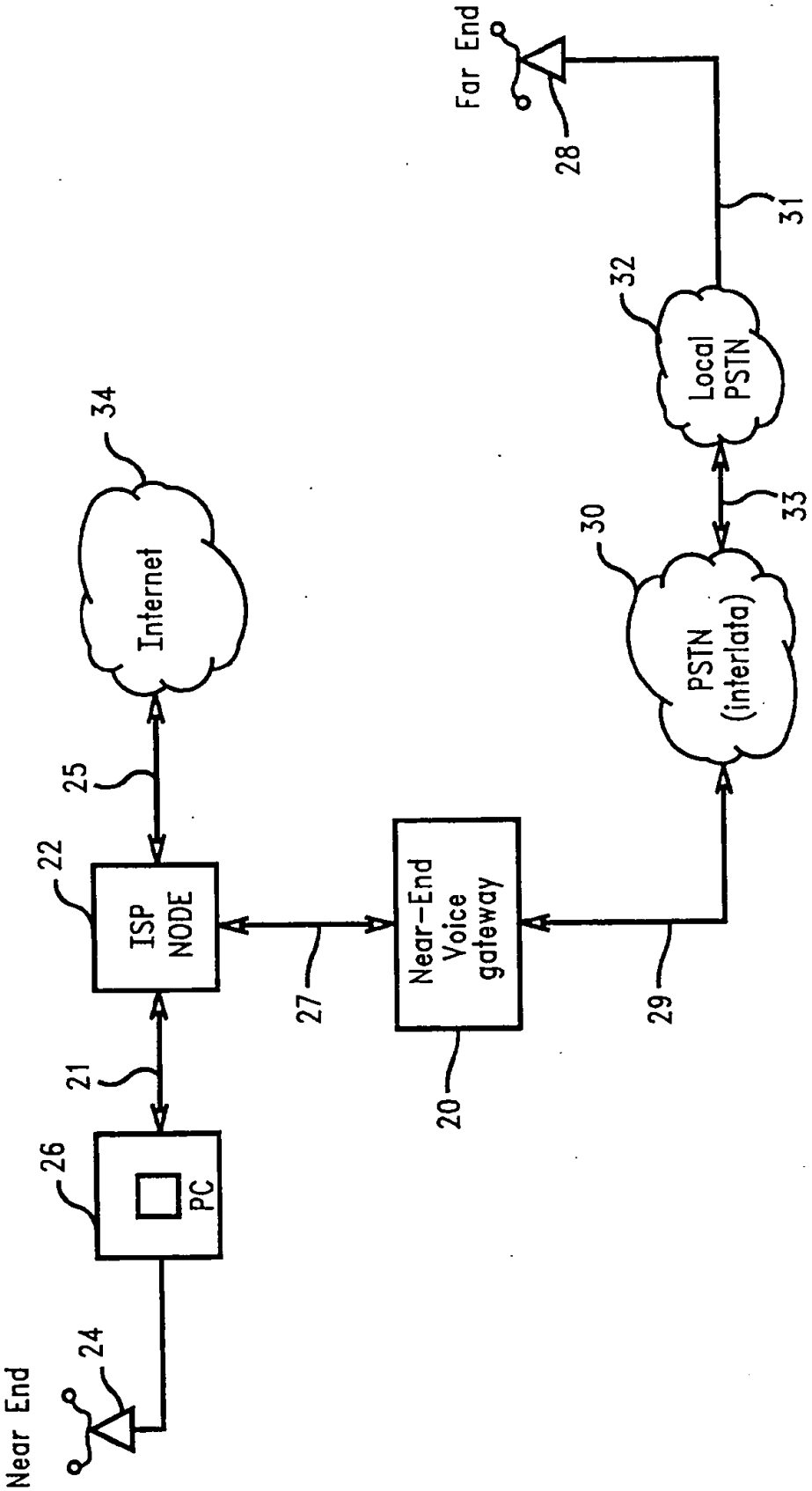
Near End
24
PC
26
21
ISP NODE
22
25
Internet
34
27
Near-End Voice gateway
20
29
PSTN (interlata)
30
33
Local PSTN
32
31
Far End
28

FIG. 1

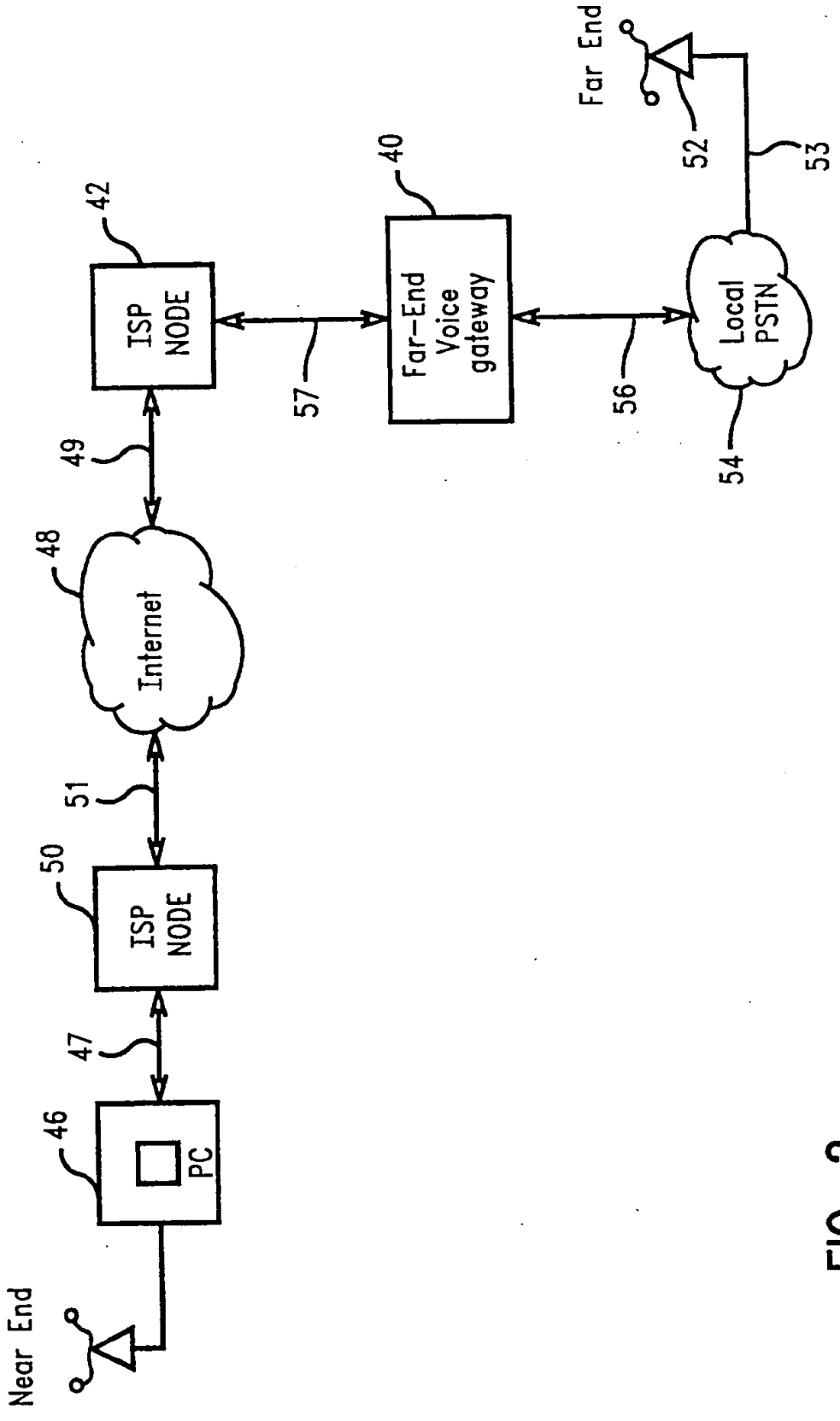
Near End
PC
46
47
ISP NODE
50
51
Internet
48
49
ISP NODE
42
57
Far-End Voice gateway
40
56
Local PSTN
54
52
Far End
53

FIG. 2

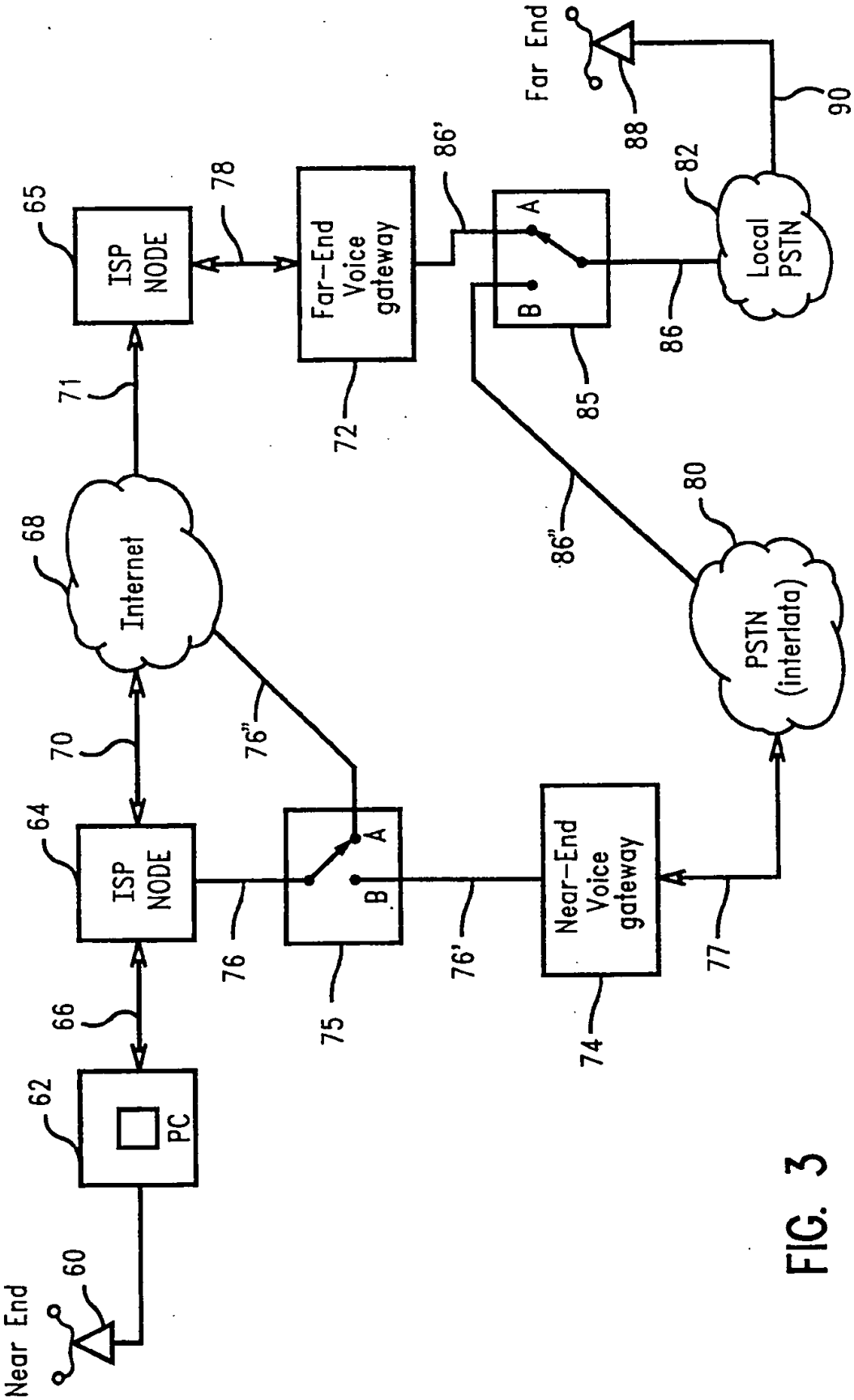
Near End
60
62
PC
66
64
ISP NODE
70
68
Internet
71
65
ISP NODE
78
Far-End Voice gateway
72
86'
A
B
85
86
Local PSTN
82
Far End
88
90
86"
80
PSTN (interlata)
76"
A
B
75
76
76'
Near-End Voice gateway
74
77

FIG. 3

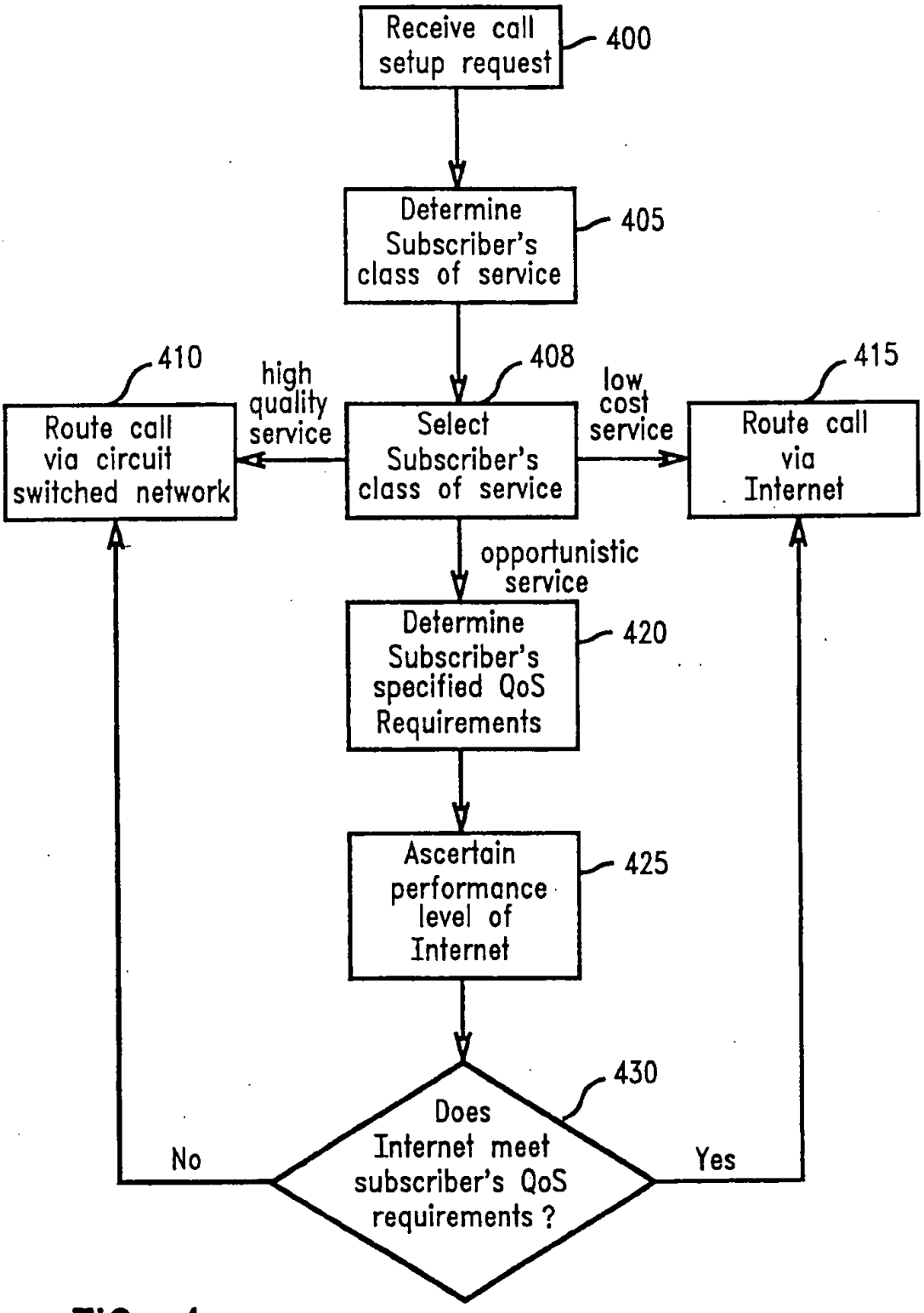
Receive call setup request
400
Determine Subscriber's class of service
405
410
high quality service
408
low cost service
415
Route call via circuit switched network
Select Subscriber's class of service
Route call via Internet
opportunistic service
Determine Subscriber's specified QoS Requirements
420
Ascertain performance level of Internet
425
430
Does Internet meet subscriber's QoS requirements ?
No
Yes

FIG. 4

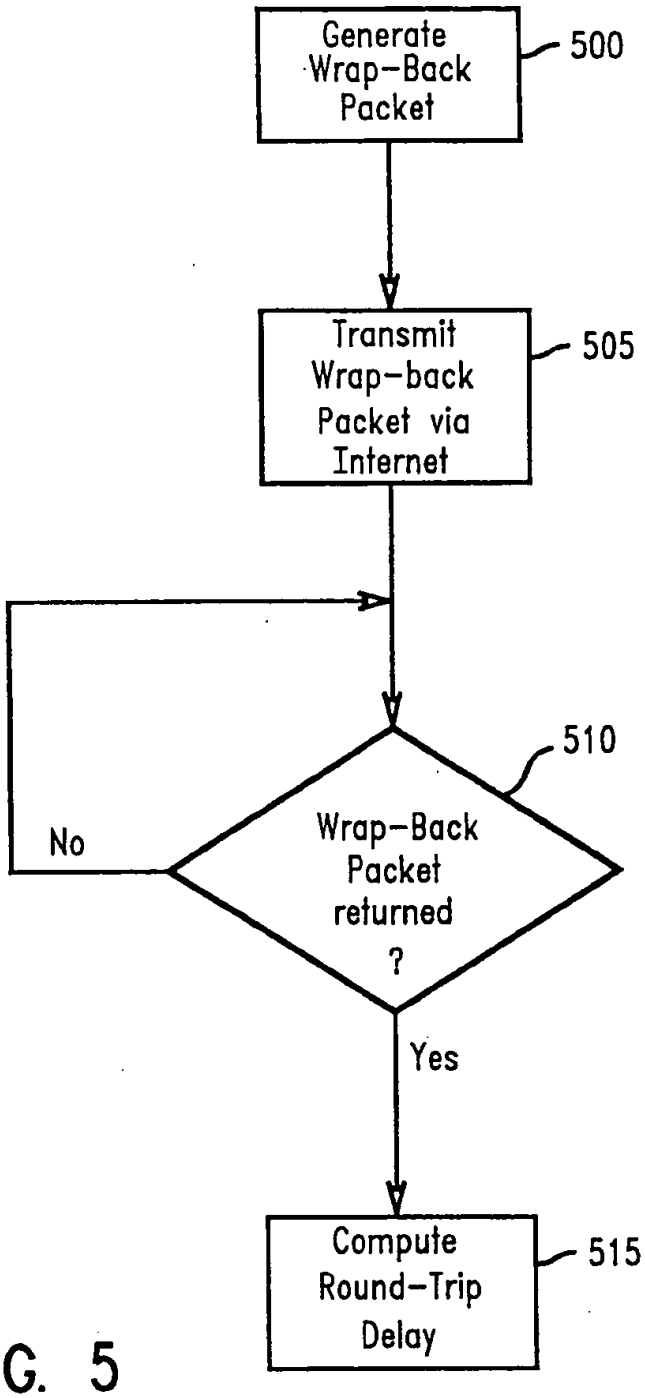
Generate Wrap-Back Packet
500
Transmit Wrap-back Packet via Internet
505
Wrap-Back Packet returned ?
510
No
Yes
Compute Round-Trip Delay
515

FIG. 5

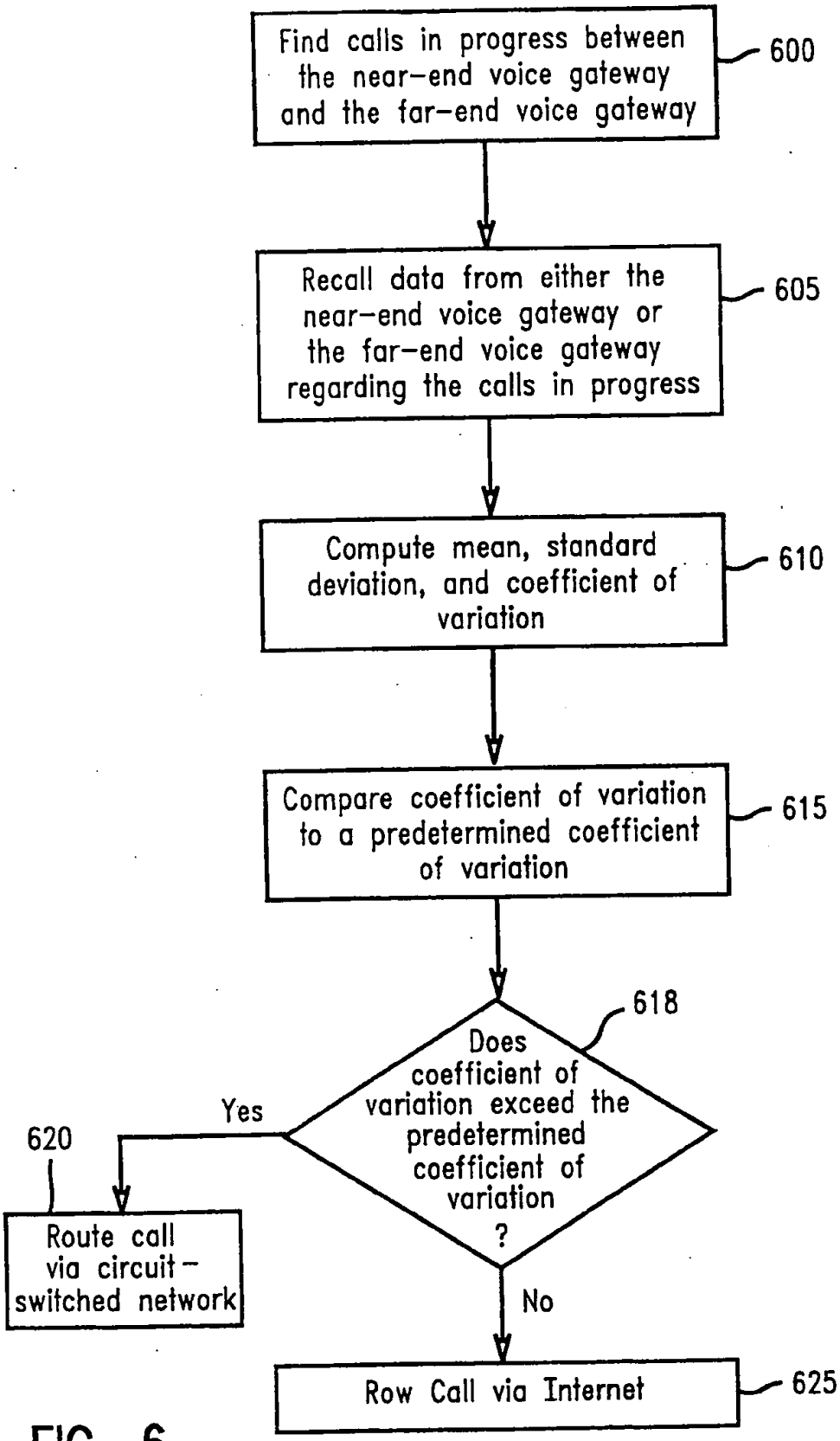
Find calls in progress between the near-end voice gateway and the far-end voice gateway
600
Recall data from either the near-end voice gateway or the far-end voice gateway regarding the calls in progress
605
Compute mean, standard deviation, and coefficient of variation
610
Compare coefficient of variation to a predetermined coefficient of variation
615
618
Does coefficient of variation exceed the predetermined coefficient of variation ?
Yes
620
Route call via circuit-switched network
No
Row Call via Internet
625

FIG. 6